ELSEVIER

Speech Communication 28 (1999) 333–334

SPEECH COMMUNICATION

www.elsevier.nl/locate/specom

Author index of Volume 28

Aguilar, L., Hiatus and diphthong: Acoustic cues and speech situation differences (1) 57–74
Alku, P., J. Vintturi and **E. Vilkman,** On the linearity of the relationship between the sound pressure level and the negative peak amplitude of the differentiated glottal flow in vowel production (4) 269–281
Arai, T., see **Kanedera, N.** (1) 43–55
Arslan, L.M., Speaker Transformation Algorithm using Segmental Codebooks (STASC) (3) 211–226
Avendano, C., see **Yegnanarayana, B.** (1) 25–42

Dogil, G., see **Jilka, M.** (2) 83–108

Furui, S., see **Ohtsuki, K.** (2) 155–166

Hanna, P., J. Ming and **F.J. Smith,** Interframe dependence arising from preceding and succeeding frames – Application to speech recognition (4) 301–312
Hermansky, H., see **Kanedera, N.** (1) 43–55
Hermansky, H., see **Yegnanarayana, B.** (1) 25–42
Hirose, K., see **Jiang, H.** (4) 313–326
Houtgast, T., see **Steeneken, H.J.M.** (2) 109–123
Hung, W.-W., see **Wang, H.-C.** (3) 243–260
Huo, Q., see **Jiang, H.** (4) 313–326

Jiang, H., K. Hirose and **Q. Huo,** Improving Viterbi Bayesian predictive classification via sequential bayesian learning in robust speech recognition (4) 313–326
Jilka, M., G. Möhler and **G. Dogil,** Rules for the generation of ToBI-based American English intonation (2) 83–108

Kanedera, N., T. Arai, H. Hermansky and **M. Pavel,** On the relative importance of various components of the modulation spectrum for automatic speech recognition (1) 43–55

Lee, S. and **Y.-H. Oh,** Tree-based modeling of prosodic phrasing and segmental duration for Korean TTS systems (4) 283–300

Maragos, P., see **Potamianos, A.** (3) 195–209
Matsuoka, T., see **Ohtsuki, K.** (2) 155–166
Ming, J., see **Hanna, P.** (4) 301–312
Minker, W., Design considerations for knowledge source representations of a stochastically-based natural language understanding component (2) 141–154
Möhler, G., see **Jilka, M.** (2) 83–108
Mohri, M. and **M. Riley,** Network optimizations for large-vocabulary speech recognition (1) 1–12
Möller, S. and **R. Schönweiler,** Analysis of infant cries for the early detection of hearing impairment (3) 175–193
Mori, T., see **Ohtsuki, K.** (2) 155–166

Oh, Y.-H., see **Lee, S.** (4) 283–300
Ohtsuki, K., T. Matsuoka, T. Mori, K. Yoshida, Y. Taguchi, S. Furui and **K. Shirai,** Japanese large-vocabulary continuous-speech recognition using a newspaper corpus and broadcast news (2) 155–166

Pavel, M., see **Kanedera, N.** (1) 43–55
Pols, L.C.W., see **van Son, R.J.J.H.** (2) 125–140
Potamianos, A. and **P. Maragos,** Speech analysis and synthesis using an AM–FM modulation model (3) 195–209

Riley, M., see **Mohri, M.** (1) 1–12

Satyanarayana Murthy, P., see **Yegnanarayana, B.** (1) 25–42
Schönweiler, R., see **Möller, S.** (3) 175–193
Shirai, K., see **Ohtsuki, K.** (2) 155–166
Smith, F.J., see **Hanna, P.** (4) 301–312
Steeneken, H.J.M. and **T. Houtgast,** Mutual dependence of the octave-band weights in predicting speech intelligibility (2) 109–123

Taguchi, Y., see **Ohtsuki, K.** (2) 155–166

van Son, R.J.J.H. and **L.C.W. Pols,** An acoustic description of consonant reduction (2) 125–140

0167-6393/99/$ – see front matter © 1999 Published by Elsevier Science B.V. All rights reserved.
PII: S0167-6393(99)00023-0

Vilkman, E., see **Alku, P.** (4) 269–281
Vintturi, J., see **Alku, P.** (4) 269–281

Wang, H.-C. and **W.-W. Hung,** Smoothing hidden Markov models by using an adaptive signal limiter for noisy speech recognition (3) 243–260
Wang, H.-C., see **Yuo, K.-H.** (1) 13–24
Wang, H.-C., see **Yuo, K.-H.** (3) 227–241

Yegnanarayana, B., C. Avendano, H. Hermansky and **P. Satyanarayana Murthy,** Speech enhancement using linear prediction residual (1) 25–42
Yoshida, K., see **Ohtsuki, K.** (2) 155–166
Yuo, K.-H. and **H.-C. Wang,** Robust features for noisy speech recognition based on temporal trajectory filtering of short-time autocorrelation sequences (1) 13–24
Yuo, K.-H. and **H.-C. Wang,** Joint estimation of feature transformation parameters and Gaussian mixture model for speaker identification (3) 227–241